IN THE SPECIFICATION:

The specification as amended below with replacement paragraphs shows added text with <u>underlining</u> and deleted text with ~~strikethrough~~.

Please REPLACE the paragraph beginning at page 3, line 2, with the following paragraph:

According to an aspect of the present invention, there is provided a wide-band speech coder, the wide-band speech coder comprising a speech characteristic classification unit, which stipulates a characteristic of speech corresponding to a current frame statistically using an open-circuit pitch value and a linear prediction coefficient in which a wide-code speech signal to be coded is perceptual ~~weigh~~<u>weight</u> filtered, an adaptive codebook retrieving unit, which retrieves a pitch delay value around the open-circuit pitch value, calculates a pitch gain value, generates an adaptive codebook contribution signal corresponding to the retrieved pitch delay value, and outputs a difference between the generated adaptive codebook contribution signal and the perceptual ~~weigh~~<u>weight</u> filtered signal as a first fixed codebook target signal, a first fixed codebook retrieving unit, which obtains a first fixed codebook index that can express the first fixed codebook target signal most properly, and a first fixed codebook gain value, generates a first fixed codebook contribution signal corresponding to the retrieved index, and outputs a difference between the first generated fixed codebook contribution signal and the first fixed codebook target signal as a second fixed codebook target signal, a second fixed codebook retrieving unit, which includes at least two second fixed codebooks according to a speech characteristic, selects a second fixed codebook according to the speech characteristic, and retrieves second fixed codebook indices that can express the second fixed codebook target signal most properly, and second fixed codebook gain values, and a parameter multiplexer, which quantizes and multiplexes the speech characteristic information, the pitch delay value, the pitch gain value, the first fixed codebook index, the first fixed codebook gain value, the second fixed codebook indices, and the second fixed codebook gain values, makes them as a bit stream, and transmits the bit stream to an external speech decoding terminal.

Please REPLACE the paragraph beginning at page 3, line 23, with the following paragraph:

According to another aspect of the present invention, there is provided a wide-band speech coding method, the wide-band speech coding method comprising (a) stipulating a characteristic of speech corresponding to a current frame statistically using an open-circuit pitch value and a linear prediction coefficient in which a wide-code speech signal to be coded is perceptual ~~weigh~~weight filtered, (b) obtaining a pitch delay value around the open-circuit pitch value and a pitch gain value and generating a difference between an adaptive codebook contribution signal corresponding to the obtained pitch delay value and the perceptual ~~weigh~~weight filtered signal as a first fixed codebook target signal, (c) obtaining a first fixed codebook index that can express the first fixed codebook target signal most properly, and a first fixed codebook gain value and generating a difference between a first fixed codebook contribution signal generated using the first obtained fixed codebook index and the first fixed codebook gain value and the first fixed codebook target signal as a second fixed codebook target signal, (d) selecting and retrieving a second fixed codebook retrieving unit from a plurality of second fixed codebooks classified according to a speech characteristic, according to speech characteristic information and retrieving second fixed codebook indices that can express the second fixed codebook target signal most properly, and second fixed codebook gain values, and (e) quantizing and multiplexing the speech characteristic information, the pitch delay value, the pitch gain value, the first fixed codebook index, the first fixed codebook gain value, the second fixed codebook indices, and the second fixed codebook gain values, making them as a bit stream, and transmitting the bit stream to an external speech decoding terminal.

Please REPLACE the paragraph beginning at page 6, line 4, with the following paragraph:

FIG. 1 is a block diagram schematically showing a wide-band speech coder according to an embodiment of the present invention. Referring to FIG. 1, the wide-band speech coder according to the embodiment of the present invention includes a pre-processing filter 101, a linear prediction coefficient analyzer 102, a perceptual ~~weighing~~weighting filter 103, an open-circuit pitch retrieving unit 104, a speech

characteristic classification unit 105, an adaptive codebook retrieving unit 106, and first and second fixed codebook retrieving units 107 and 108.

Please REPLACE the paragraph beginning at page 6, line 13, with the following paragraphs:

The linear prediction coefficient analyzer 102 analyzes a linear prediction coefficient of the signal pre-processed by the pre-processing filter 101 to obtain the linear prediction coefficient and embodies the perceptual ~~weighing~~weighting filter 103 for use in codebook retrieving using the linear prediction coefficient.

The perceptual ~~weighing~~weighting filter 103 ~~weighs~~weights quantization noise of an auditorily sensitive frequency wide-band and performs perceptual ~~weighing~~weighting filtering the pre-processed signal so that efficient coding is performed.

The open-circuit pitch retrieving unit 104 performs open-circuit pitch retrieving using the signal that is perceptually ~~weigh~~weight filtered by the perceptual ~~weighing~~weighting filter 103.

Please REPLACE the paragraph beginning at page 6, line 29, with the following paragraph:

The adaptive codebook retrieving unit 106 retrieves an adaptive codebook 106a using the open-circuit pitch value obtained by the open-circuit pitch retrieving unit 104 through open-circuit pitch retrieving. The adaptive codebook 106a is composed of a pitch delay value and a pitch gain value. The adaptive codebook retrieving unit 106 retrieves the pitch delay value around the open-circuit pitch value obtained by the open-circuit pitch retrieving unit 104, and simultaneously calculates the pitch gain value to output the pitch gain value and the pitch delay value to a parameter multiplexer 110. In addition, the adaptive codebook retrieving unit 106 generates an adaptive codebook contribution signal corresponding to the retrieved pitch delay value and outputs a difference between the generated adaptive codebook contribution signal and the speech signal output from the perceptual ~~weighing~~weighting filter 103 as a first fixed codebook target signal to the first fixed codebook retrieving unit 107.

Please REPLACE the paragraph beginning at page 10, line 1, with the following paragraph:

A signal output from the multiplier 311 goes through an adaptive pre-processing filter 302 and a perceptual ~~weighing~~weighting synthesis filter 303 considering a periodic characteristic of speech, thereby generating a fixed codebook contribution signal. A subtracter 309 subtracts a fixed codebook contribution signal from the fixed codebook target signal and obtains an index and a gain value for a code having a smallest error between the fixed codebook target signal and the fixed codebook contribution signal, as a subtraction result, as a first fixed codebook index and a first fixed codebook gain value. In this case, a mean square error (MSE) system may be used for error measurement. Retrieving of an optimum fixed codebook may be obtained using Equation 1.

Please REPLACE the paragraph beginning at page 10, line 14, with the following paragraphs:

In this case, $d_1(n)$ is a target signal of a first fixed codebook, G_{c1} is a first fixed codebook gain value, $h_w(n)$ is a shock response signal of a perceptual ~~weighing~~weighting filter, and $c_1(n)$ is a first fixed codebook pulse signal.

As shown in FIG. 3, one of an algebraic codebook 304 and a random codebook 305 may be selected according to a speech characteristic and may be used as the second fixed codebook. As described previously, in a speech section where a noise characteristic is strong, such as fricative sound or affricate, or in an unvoiced sound section, a random codebook may be used as the second fixed codebook, and in other sections, an algebraic codebook may be used as the second fixed codebook. The multiplier 306 multiplies the code vector retrieved by the algebraic codebook 304 or the random codebook 305 by a second fixed codebook gain value. Here, the second fixed codebook gain value, as described previously, is calculated using the index of the retrieved code vector and the second fixed codebook target signal. Like in retrieving of the first fixed codebook, a signal output from the multiplier 306 goes through an adaptive pre-processing filter 307 and a perceptual ~~weighing~~weighting synthesis filter 308, thereby generating a second fixed codebook contribution signal. And, a subtracter 310 subtracts a second fixed codebook contribution signal from the second fixed codebook target

signal and obtains an index and a gain value for a code having a smallest error between the second fixed codebook target signal and the second fixed codebook contribution signal, as a subtraction result, as a second fixed codebook index and a second fixed codebook gain value. In this case, a mean square error (MSE) system may be used for error measurement.